Speech Enhancement using DWT for Hearing Aid Application

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Abstract—Hearing impairment is the number one chronic disability affecting people in the world. With a loss of hearing, person is restricted from his or her normal social activity, which may in turn cause undesirable influence on mental health. An effective treatment for the problem is hearing assistive device called hearing aid. With reference to patient's audiogram hearing aid device would enhance signal such that hearing impaired person can hear comfortably. It has been tried to implement same technical concept. Presented approach for speech enhancement for hearing aid by using discrete wavelet transform-multiresolution analysis.
Keywords- *DWT; IDWT; STFT; FPGA; Wavelet Transform; MRA; DPHA*

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I. Introduction

For human being, 'hearing' is more than a mere sensory mechanism. With a loss of hearing, person is restricted from his or her normal social activity, which may in turn cause undesirable influence on mental health. The most common cause of impaired hearing is old age and long-term exposure of the ear to sound energy [11]. An effective treatment for the problem is hearing assistive device called hearing aid. A hearing aid is a small electronic device that one wears in or behind his/her ear. The main function of a hearing-aid device is to amplify those sound in which patient has hearing loss, and then transfer the processed signal to the patient's ear. So Hearing aid enhance the signal up to the level such that hearing impaired person can hear comfortably.

Modern hearing aids can be divided into three types: digital hearing and analog hearing aid. Analog hearing aids are the low cost hearing aid. Their limitations is that it gives user independent response and also it amplifies sound without distinguishing among different sounds in noisy environment.

Most of the hearing aid market are captured by Digital hearing aids. They are more preferable over analog hearing aids because of using advanced digital signal processing (DSP) algorithms to compensate speech signal and improve intelligibility in noisy environment, as well as echo or feedback cancellation [6]. There are a number of techniques for the hearing aid: fourier transform, filter banks, wavelet transform. Among them discrete wavelets transform provides good frequency resolution and therefore used here for speech enhancement [7]. The number of dB by which the threshold sound pressure of an individual exceeds the normal threshold is referred to as dB of "hearing loss". Figure 1 represents the most common hearing loss (due to aging).

II. Techniques for Speech Enhancement

There are different techniques for the hearing aid: fast fourier transform, short term fourier transform, uniform/non-uniform filter bank, wavelet transform.

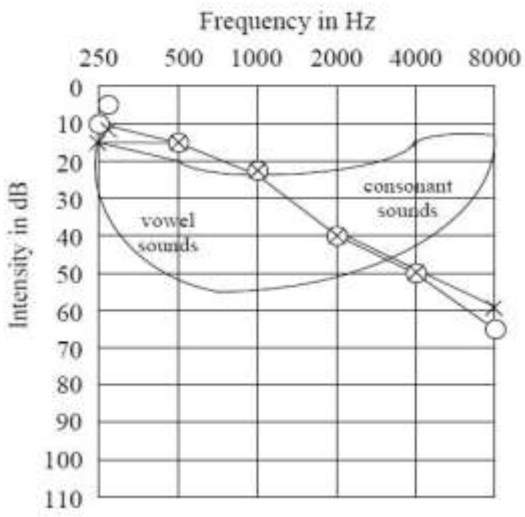


Figure 1. Audiogram with hearing loss [11]

The Fourier transform techniques gives frequency information of speech signal, but it does not give information about at what time that frequency components was present. So fourier transform is not suitable for non-stationary signals. But most of the real time signals are non-stationary signals. While short term fourier transform (STFT) uses fixed size window and gives information about both time and frequency of signal. Filter banks uses parallel of low pass, band pass and high pass filter and divide the signal into different frequency bands and performs the speech enhancement as per the requirement from the patient's audiogram. While discrete wavelet transform performs richer frequency resolution as compared to previous techniques by dilation and translation of wavelet. Discrete Wavelet Transform represents the signal in time and frequency domain. So, it is suitable for non-stationary real time signals like speech.

III. Discrete Wavelet Transform and Multi Resolution Analysis

DWT decompose the signal into multiple stages. DWT uses two functions mother wavelet function and approximation function. Mathematically wavelet is denoted by equation – 1.

$$\psi_{a,b} = \frac{1}{\sqrt{a}}\psi\left(\frac{t-b}{a}\right) \quad \text{--- (1)}$$

Where,

the variable “b” represents time shift or translation
the variable “a” represents scale or dilation variable
the function ψ(t) represents mother wavelet

Scaling of wavelet gives high frequency wavelet and dilation of wavelet gives low frequency wavelet. By dilation and compression of wavelet it generates child wavelet. This wavelet is convolve with speech signal. If high similarities exist between wavelet and signal, generates high valued wavelet coefficients and which indicates existence of that frequency. DWT decompose the signals into different levels. Every level has unique time and frequency resolution. Figure 2 shows 3-stage DWT. The output of wavelet function blocks gives detail component of signal, also called high frequency component of signal. The output of approximation function gives low frequency component of signal. Both the high and low frequency coefficients are down sampled by two. And these down sampled approximation coefficients passes to next stage. So, output of three stage DWT is three detail coefficients and one approximate coefficients.

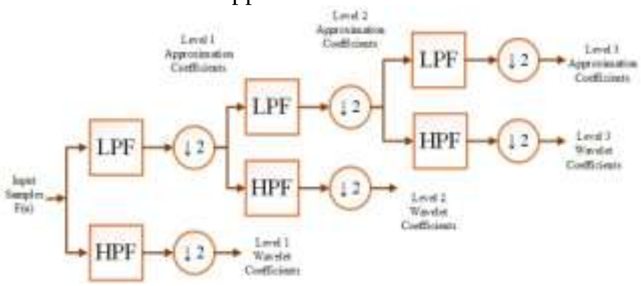


Figure 2. Three stages in Forward Discrete Wavelet Transform decomposition

IV. SPEECH ENHANCEMENT ALGORITHM

Algorithm to enhance the speech signal for hearing aid is shown in figure below. Input digitalized speech samples should be in power of two. And it is given as a input to the DWT block. Numbers of level of decomposition of samples depends on the requirement of frequency resolution. DWT gives wavelet coefficients at different stages. These wavelet coefficients multiplied with some value as per requirement from the patient’s audiogram. These multiplicand value depends on the intensity of incoming signal and hearing threshold value of patient’s at that frequency. The signal is recovered from the modified wavelet transform by applying inverse discrete wavelet transform (IDWT).

V. SIMULATION RESULTS

Speech enhancement using DWT is implemented on MATLAB which divide signals into nine discrete bands of frequencies. Daubechies wavelet with four nonzero coefficients (DB4) is used. The different .wav file including men voice, music, bird voice, song with features listed in Table 1 is applied to this block. Then individual band is processed and enhanced with different value of multiplied coefficients. Increment loudness level in decibel in individual is observed.

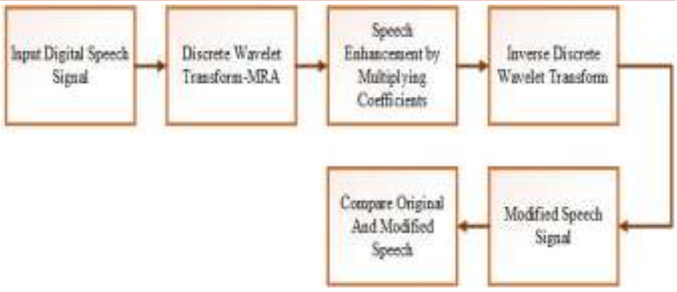


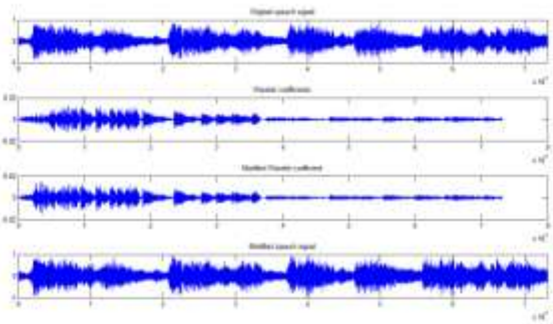
Figure 3. Implemented speech enhancement algorithm

Table 1. Properties of Speech Files

Audio Signal	Fs	Nbits	Format	Duration
Man voice(1)	8000	16	PCM	2
Man voice(2)	11025	8	PCM	8.3
Music	8192	16	PCM	8.9
Bird	8192	16	PCM	1.6
Song	44100	16	PCM	8
Sample	8192	16	PCM	5

In table-1, ‘Fs’ is sampled frequency of audio samples and ‘Nbits’ is number of bits per sample.

Speech signal: music signal
Speech enhancement in fifth band of frequency.
Value of multiplied coefficient: 3
Average increment in fifth band of frequency: 13 db



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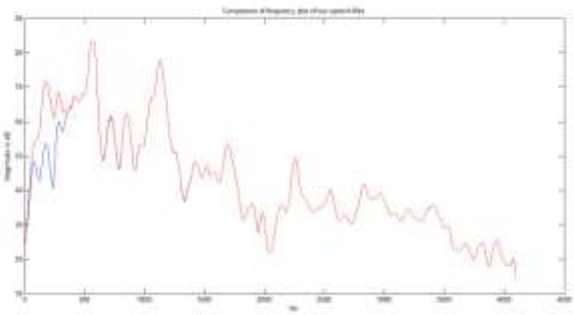

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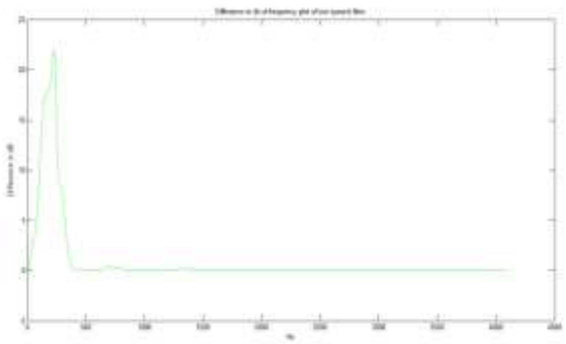
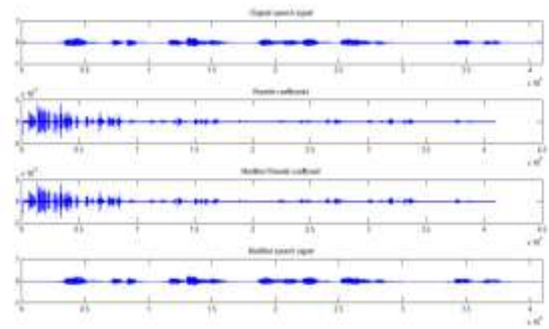


(c)

Figure 4. Music signal waveform- (a) original, wavelet coefficients, modified wavelet coefficients, modified speech signal, (b), frequency spectrum of original signal and modified signal, (c) difference of two speech signals in dB

Speech signal: Recorded sample audio
Speech enhancement in ninth band of frequency.
Value of multiplied coefficient: 1.25
Average increment in ninth band of frequency: 2.5 db



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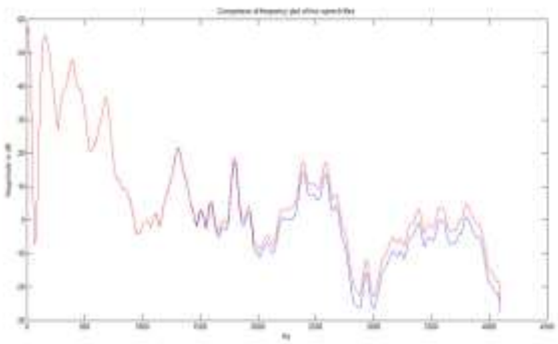

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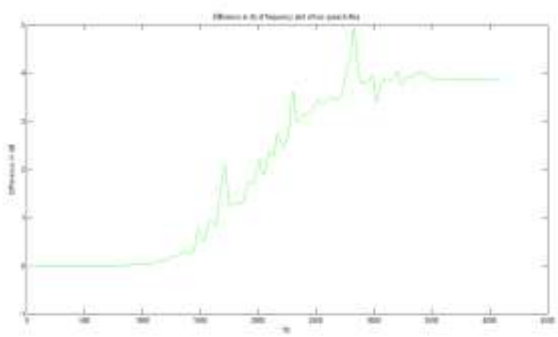

(c)

Figure 5. Recorded signal waveform- (a) original, wavelet coefficients, modified wavelet coefficients, modified speech signal, (b), frequency spectrum of original signal and modified signal, (c) difference of two speech signals in dB

Table 2 shows simulation results for different applied input signals. In Table 2 “X” is value of multiplied coefficients.

Table 2. Simulation results

Audio Signal	Freq. Band	X	Average Increment(dB)
Man Voice(1)	1st	1.6	7
	2nd	1.9	0.9
	4th	1.7	7
	7th	1.3	2.8
	9th	1.2	2.1
Man Voice(2)	1st	2.1	10
	3rd	2.3	8.2
	5th	2.35	8.23
	8th	1.76	6
Music	2nd	2	3.1
	5th	3	13
	7th	1.4	3.5
	9th	2	10
Bird	3rd	2.5	5.8
	6th	2.8	13.5
	7th	2.25	23
Song	1st	1.6	6.5
	4th	1.7	5.5
	6th	1.3	2.6
Sample	3rd	2.5	12
	6th	1.8	6.2
	9th	1.25	2.5

From simulation results it has been noticed that signal enhancement from 2 dB to 13 dB for various speech signal having different frequency is achieved. For different signals, selective band enhancement with require dB is achieved.

VI. PROPOSED HARDWARE IMPLEMENTATION

Now days, implementing soft processor on FPGA chip and implementing DSP module on FPGA and then combining it with soft processor as a peripheral are employed which is called System on Programmable Chip (SOPC). FPGA with soft processor is good choice for programmable hearing aid this is because it run faster than DSP chip does, due to of the strong ability of FPGA of parallel processing [1]. Example of one of the soft processor is NIOS-II from Altera. The proposed implementation platform for Digital Programmable Hearing Aid (DPHA) is shown in Figure 4. Programmable means user can adjust the value of multiplication coefficients as per the user’s audiogram need.

A. Speech Processing Flow

The flow of speech processing of proposed programmable hearing aid is shown in Figure 7. The input speech signal coming from the microphone is first amplified and then transferred to ADC of the audio CODEC. The ADC of audio CODEC sample the analog signal and convert it into digitalized samples. Then this samples are transmitted to store into the RAM of the FPGA board. These received data from memory is transmitted to the DWT. As per the audiogram of patient particular band of wavelet coefficients are multiplied with constant. To produce the digitalized signal back to time domain, the Inverse DWT block is activated by the soft processor. The IDWT uses updated new wavelet coefficients. These processed samples are stored to RAM. Then the DAC of the audio codec read this modified digital samples and convert this samples into analog signal back.

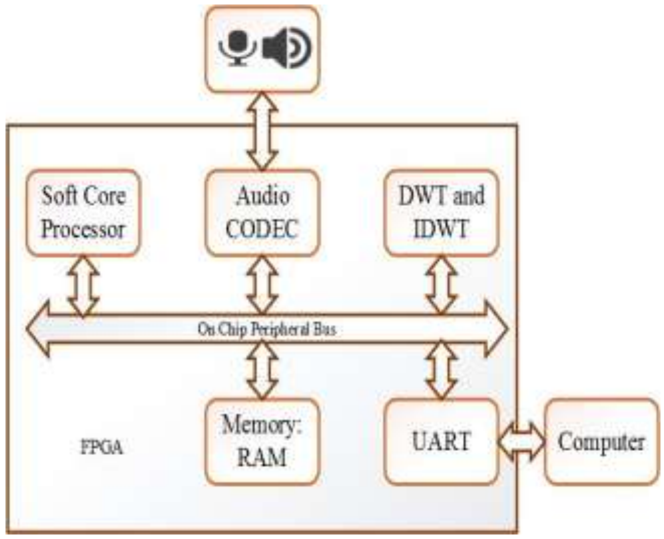

Figure 6. Block diagram of Hardware Implementation Platform for DPHA

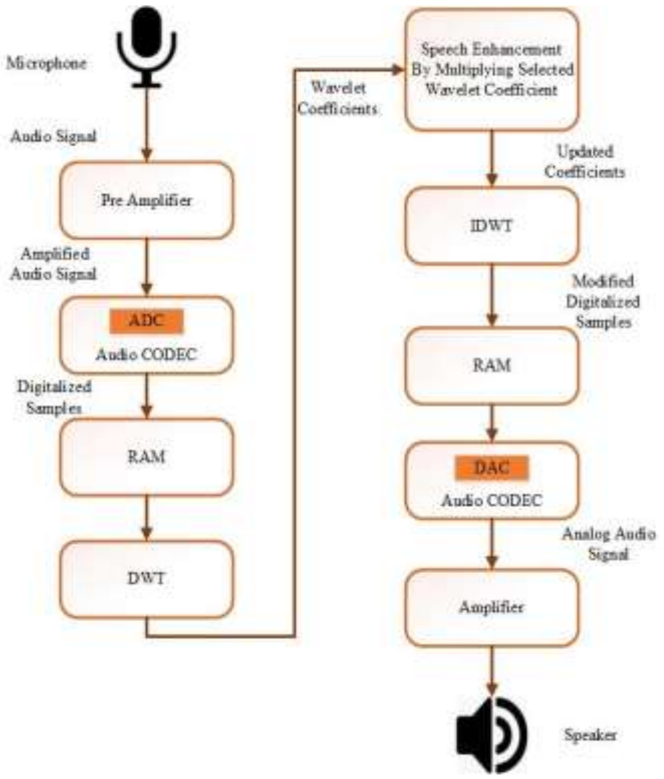

Figure 7. Flow of speech processing

VII. CONCLUSION

The main motivation behind using the wavelet analysis for speech processing in hearing aid is, wavelets have good localization in time and frequency domain. Wavelet transforms is computationally less complex with concept of multiresolution approximation. So by applying wavelet with MRA, speech signal is divided in 9-frequency bands. Wavelets coefficients are available for each and every corresponding band. As per the requirement, by multiplying coefficients of wavelets with suitable numerical values before reconstructing the signal it is possible to get increment in decibels as shown in the results. For selective frequency band of signals 2 dB to 13 dB average enhancement has been obtained. Implementing programmable hearing aid using FPGA with soft processor will give real time speech processing for hearing aid.

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